Claims

1. A method for processing an audio signal received through a microphone array, comprising:

receiving a signal;

applying adaptive beam-forming to the signal to yield an enhanced source component of the signal;

applying inverse beam-forming to the signal to yield an enhanced noise component of the signal; and

combining the enhanced source component and the enhanced noise component to produce a noise reduced signal.

2. The method of claim 1, wherein the method operation of combining the enhanced source component and the enhanced noise component to produce a noise reduced signal includes,

aligning the enhanced noise component of the signal through an adaptive filter.

3. The method of claim 1, further comprising:

canceling acoustic echoes from the signal.

4. The method of claim 1, wherein the method operation of applying adaptive beam-forming to the signal to yield an enhanced source component of the signal includes,

enhancing a broadside noise signal;

calculating a calibration coefficient;

applying the calibration coefficient to the enhanced broadside noise signal; and

adjusting a listening direction based upon the calibration coefficient.

5. The method of claim 1, wherein the method operation of applying adaptive beam-forming to the signal to yield an enhanced source component of the signal includes,

analyzing the signal; and

separating the signal into a noise component signal and a source signal.

6. The method of claim 5, wherein the method operation of separating the signal into a noise component signal and a source signal includes,

calculating second order statistics associated with the signal.

7. A method for reducing noise associated with an audio signal received through a microphone sensor array, comprising:

enhancing a target signal component of the audio signal through a first filter;

blocking the target signal component through a second filter;

combining an output of the first filter and an output of the second filter in a manner to reduce noise without distorting the target signal;

periodically monitoring an acoustic set-up associated with the audio signal; and

calibrating both a value of the first filter and a value of the second filter based upon the acoustic set-up.

8. The method of claim 7, further comprising:

defining the target signal component and a noise signal component through second order statistics.

9. The method of claim 8, further comprising:

separating the target signal component and the noise signal component; and

determining a time delay associated with each microphone sensor of the microphone senor array.

10. The method of claim 7, wherein the method operation of combining the output of the first filter and the output of the second filter in a manner to reduce noise without distorting the target signal includes,

aligning the output of the second filter.

11. The method of claim 7, wherein the acoustic set-up refers to relative position of a user and the microphone sensor array.

12. The method of claim 7, wherein the method operation of periodically monitoring an acoustic set-up associated with the audio signal includes occurs about every 100 milliseconds.

13. The method of claim 7, wherein the method operation of calibrating both a value of the first filter and a value of the second filter based upon the acoustic set-up includes,

applying a blind source separation scheme using second order statistics associated with the audio signal.

14. A computer readable medium having program instructions for processing an audio signal received through a microphone array, comprising:

program instructions for receiving a signal;

program instructions for applying adaptive beam-forming to the signal to yield an enhanced source component of the signal;

program instructions for applying inverse beam-forming to the signal to yield an enhanced noise component of the signal; and

program instructions for combining the enhanced source component and the enhanced noise component to produce a noise reduced signal.

15. The computer readable medium of claim 14, wherein program instructions for combining the enhanced source component and the enhanced noise component to produce a noise reduced signal includes,

program instructions for aligning the enhanced noise component of the signal through an adaptive filter.

16. The computer readable medium of claim 14, further comprising:

program instructions for canceling acoustic echoes from the signal.

17. The computer readable medium of claim 14, wherein the program instructions for applying adaptive beam-forming to the signal to yield an enhanced source component of the signal includes,

program instructions for enhancing a broadside noise signal;

program instructions for calculating a calibration coefficient;

program instructions for applying the calibration coefficient to the enhanced broadside noise signal; and

program instructions for adjusting a listening direction based upon the calibration coefficient.

18. The computer readable medium of claim 14, wherein the program instructions for applying adaptive beam-forming to the signal to yield an enhanced source component of the signal includes,

program instructions for analyzing the signal; and

program instructions for separating the signal into a noise component signal and a source signal.

19. The computer readable medium of claim 18, wherein the program instructions for separating the signal into a noise component signal and a source signal includes,

program instructions for calculating second order statistics associated with the signal.

20. A computer readable medium having program instructions for reducing noise associated with an audio signal, comprising:

program instructions for enhancing a target signal associated with a listening direction through a first filter;

program instructions for blocking the target signal through a second filter;

program instructions for combining an output of the first filter and an output of the second filter in a manner to reduce noise without distorting the target signal;

program instructions for periodically monitoring an acoustic set up associated with the audio signal; and

program instructions for calibrating both the first filter and the second filter based upon the acoustic setup.

21. The computer readable medium of claim 20, further comprising:

program instructions for defining the target signal component and a noise signal component of the audio signal through second order statistics.

22. The computer readable medium of claim 21, further comprising:

program instructions for separating the target signal component and the noise signal component; and

program instructions for determining a time delay associated with each microphone sensor of the microphone senor array.

23. The computer readable medium of claim 20, wherein the program instructions for combining the output of the first filter and the output of the second filter in a manner to reduce noise without distorting the target signal includes,

program instructions for aligning the output of the second filter.

24. The computer readable medium of claim 20, wherein the program instructions for calibrating both a value of the first filter and a value of the second filter based upon the acoustic set-up includes,

program instructions for applying a blind source separation scheme using second order statistics associated with the audio signal.

25. A system capable of isolating a target audio signal from multiple noise sources, comprising:

a portable consumer device configured to move independently from a user;

a computing device, the computing device including logic configured enhance the target audio signal without constraining movement of the portable consumer device; and

a microphone array affixed to the portable consumer device, the microphone array configured to capture audio signals, wherein a listening direction associated with the microphone array is controlled through the logic configured to enhance the target audio signal.

26. The system of claim 25, wherein the computing device is contained within the portable consumer device.

27. The system of claim 26, wherein the computing device includes,

logic for blocking the target signal through a second filter;

logic for combining the output of the first filter and the output of the second filter in a manner to reduce noise without distorting the target signal;

logic for periodically monitoring an acoustic set up associated with the audio signal; and

logic for calibrating both the first filter and the second filter based upon the acoustic setup.

28. The system of claim 25, wherein the microphone array is configured in one of a convex geometry and a straight line geometry.

29. The system of claim 25, wherein a distance between microphones of the microphone array is about 2.5 centimeters.

30. The system of claim 25, wherein the portable consumer device is a video game controller and the computing device is a video game console.

31. A video game controller, comprising:

a microphone array affixed to the video game controller, the microphone array configured to detect an audio signal that includes a target audio signal and noise;

circuitry configured to process the audio signal; and

filtering and enhancing logic configured to filter the noise and enhance the target audio signal as a position of the video game controller and a position of a source of the target audio signal change, wherein the filtering of the noise is achieved through a plurality of filter-and-sum operations.

32. The video game controller of claim 31, wherein the filtering and enhancing logic includes,

separation filter logic configured to separate the target audio signal from the noise through a blind source separation scheme.

33. The video game controller of claim 32, wherein the blind source separation scheme is associated with a second order statistic derived from data corresponding to the audio signal.

34. The video game controller of claim 32, wherein the separation filter logic includes,

adaptive array calibration logic configured to periodically calculate a separation filter value, the separation filter value capable of adjusting a listening direction associated with the microphone array.

35. An integrated circuit, comprising:

circuitry configured to receive an audio signal from a microphone array in a multiple noise source environment;

circuitry configured to enhance a listening direction signal;

circuitry configured to block the listening direction signal;

circuitry configured to combine the enhanced listening direction signal and the blocked listening direction signal to yield a noise reduced signal; and

circuitry configured to adjust a listening direction according to filters computed through an adaptive array calibration scheme.

36. The integrated circuit of claim 35, wherein the adaptive array calibration scheme applies a second order statistic to data associated with the audio signal to derive one of a signal passing filter and a blocking filter.

37. The integrated circuit of claim 35, wherein the adaptive array calibration scheme is periodically invoked.

38. The integrated circuit of claim 35, wherein the circuitry configured to combine the enhanced listening direction signal and the blocked listening direction signal to yield a noise reduced signal includes,

circuitry configured to align the enhanced listening direction signal with the blocked listening direction signal.

39. The integrated circuit of claim 35, wherein the integrated circuit is contained within one of a video game controller and a video game console.